METHOD AND SYSTEM FOR DYNAMIC GATEWAY SELECTION IN AN IP TELEPHONY NETWORK

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BACKGROUND OF THE INVENTION

1. Field of the Invention

The present invention relates generally to the field of IP telephony, and more particularly to a method and system for selecting gateway(s) for routing calls through a packet-based telecommunications network interconnected with a public telecommunications network.

2. Acronyms

The written description herein uses a number of acronyms to refer to various services, messages and system components. Although generally known, use of several of these acronyms is not strictly standardized in the art. For purposes of this discussion, several acronyms will be defined as follows:

Dynamic Gateway Selection (DGS) - a process employed by a network server and a Redirect Server (RS) to allow calls to be routed to an egress gateway.

Egress (Destination) and Ingress (Origination) Gateways - gateways connecting an IP network to a PSTN, PBX or other network .

Network Management System (NMS) - performs a variety of functions, including receiving and storing status changes to destination or egress gateways.

3. Description of the Prior Art

Internet telephony provides real-time delivery of voice, and other multimedia data, between two or more parties across a network employing Internet protocols (IP). Internet telephony is session-based rather than connection-based. That is, Internet telephony uses virtual connections or circuits to pass data between two nodes. These connections between the nodes are termed "virtual circuits" to distinguish them from dedicated circuits of conventional networks.

While IP telephony offers benefits to both users and carriers in terms of costs and variety of media types that can be routed threrethrough, there are a substantial amount of traditional telephones being serviced by the Public Switched Telephone Network (PSTN). The PSTN provides users with dedicated, end-to-end circuit connections for the duration of each call. Circuits are reserved between an originating switch and a terminating switch based on a called party number.

However, because of the popularity of the Internet, many public telecommunications networks now carry significantly more IP data traffic than voice data traffic. Public telecommunications networks, optimized for voice traffic, are ill-

equipped to handle increasing data traffic volumes. The growth in IP data traffic coupled with customer demands for integrated voice and data services at lower costs has led to the adoption of IP as the preferred protocols for carrying both voice and data traffic between originating and terminating switches of the public telecommunications network.

Accordingly, in order for IP based telephony services to become broadly accepted by users of traditional telephones, it is necessary to interface the IP telephony network with the existing PSTN and with private PBX phone networks. To permit this mode of operation, a packet-based network, such as the Internet, must interface directly with public telephone networks and operate as a bridge between originating and destination switches of such networks.

Media streams which originate from a public network must be capable of being transported through the packet-switched IP network and terminate at the same or different public network. This type of interfacing is performed by gateways which interface the signaling and bearer paths between the two networks. Therefore, Internet gateways perform code and protocol conversion between two otherwise incompatible networks to provide links necessary to send packets between the two different networks and thus make network-to-network connections possible. To assure overall system reliability it is crucial that the gateways are reliable and their availability, especially

destination or egress gateways, be monitored for quickly and efficiently selecting an available gateway.

SUMMARY OF THE INVENTION

In accordance with the principles of the invention, a method and system are provided by which an egress (i.e., destination) gateway is dynamically selected to establish a communication session over a path supported at least in part by an IP telephony network and a PSTN, PBX or other network. A redirect server (RS) in concert with a network management system (NMS) employs a gateway selection methodology which includes recording egress gateways which are not available due to several reasons, such as having timed out, or having a malfunction, i.e., failure status.

In one embodiment of the present invention, a method is provided for dynamically selecting an egress gateway to allow a call to be completed in a communication session over a path supported at least in part by an IP telephony network and a PSTN, PBX or other network. The IP telephony network includes a plurality of ingress and egress gateways, at least one Session Initiation Protocol (SIP) proxy server (SPS) and at least one redirect server (RS). A dynamic gateway selection (DGS) feature is always active and is typically invoked whenever a source user agent (SUA) initiates a call attempt by sending a session participation request to the SPS.

The preferred method generally includes the steps of: receiving a call setup request at the SPS from the SUA. The SPS forwards the request to the RS to obtain information of destination gateways. The RS responds to the SIP session participation request with either a redirection response or a request failure response. The RS redirection response includes a routing list. The routing list is a list of egress (i.e., destination) gateways that are determined to be in-service. Upon receipt of a redirection response from the RS, the SPS proxies the session participation request to a first destination gateway in the routing list. Otherwise, the RS returns a failure response which is sent to the SUA. The failure response is an indication that there are no destination gateways having an in-service status.

When the SPS proxies the request to a destination gateway, the SPS waits for a final response from the selected destination gateway. The SPS will either receive a final response or time-out. When the SPS receives a final response from the destination gateway, the SPS proxies the final response to the SUA, awaits an acknowledgment and proxies the acknowledgment. Otherwise, when the SPS times-out waiting for a final response from the destination gateway, the SPS re-sends the request a predetermined number of times. If the SPS times-out for a final time, the SPS sends the session participation request to the next destination gateway in the routing list provided by the RS.

The process of sending a request a predetermined number of times is repeated for the next destination gateway in the routing list until one of the following occurs: (1) a success response is returned; (2) the SPS times-out waiting for a final response, as described above with respect to the first destination gateway in the routing list; (3) the SPS receives an unsuccessful final, non-server failure response from the currently selected destination gateway; or (4) the destination gateway returns a server failure response, in which case the SPS informs the RS of the destination gateway failure.

In case of the second to fourth situations, the SPS sends the session participation request to other destination gateways in the routing list provided by the RS. For each destination gateway that returns a gateway failure response to the SPS, the destination gateway is recorded as an out-of-service destination gateway in the RS and is dynamically removed from the routing list. Therefore, subsequent requests are not sent to destination gateways which are recorded as out-of-service destination gateways in the RS until after a predetermined amount of time. After the predetermined amount of time has elapsed, the RS automatically recovers failed or out-of-service pathways and issues a report to a network management system (NMS) indicating that the destination gateway is back in service. Table structures, stored within the RS, are updated on a real-time basis when it is determined that a gateway is out-of-service or back in-service. When the session participation request has been sent to all destination

gateways and no successful response is received, the SPS returns a final response to the originating agent or calling party indicating that a call cannot be made.

In an alternative method, availability of destination gateways is determined using a ping method. In this method, gateway availability is determined by sending at least one packet to each destination gateway to ascertain whether the destination gateway is available, or in-service. If the destination gateway is in-service, it transmits an ACK message. If an ACK message is not received after a predetermined period of time, the destination gateway (DGW) is determined to be unavailable. The ping method preferably queries each destination gateway one-by-one and updates gateway information tables by recording the status of each destination gateway.

The present invention thereby provides several functions, including: (1) dynamic detection of failed and unavailable gateways; (2) dynamic removal of failed and unavailable gateway(s) from a routing list, gateway information table, etc., after a predetermined period of time; and (3) automatic recovery of failed and unavailable gateways by updating gateway status tables after a predetermined period of time.

BRIEF DESCRIPTION OF THE DRAWINGS

Various preferred embodiments are described herein with reference to the accompanying drawings, in which:

FIG. 1 is a block diagram of a preferred embodiment of the present invention;

FIG. 2 is a flowchart illustrating signaling and call setup procedures according to the embodiment of FIG.1;

FIG. 3 is a call flow diagram illustrating the signaling and call setup procedures according to the embodiment of FIG. 1; and

FIG. 4 is a flowchart illustrating an alternate embodiment of the present invention.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENTS

Preferred embodiments of the present invention will now be described with reference to the figures where like reference numbers indicate identical or similar elements. It will be apparent to persons skilled in the relevant art that the present invention can operate on many different types of networks without departing from the scope of the present invention. In the preferred embodiments described herein, the IP telephony network is preferably the Internet. Other examples of networks which could be used include leased lines, frame delay, and asynchronous transfer mode (ATM) networks.

The present invention provides a method and system for selecting an egress or destination gateway to establish a communication session over a path supported at least in part by an IP telephony network, such as a WAN, and a PSTN, PBX or other network. The method further determines the status of a destination gateway,

particularly, as being either in-service or out-of-service, and automatically brings a destination gateway back into service from an out-of-service state after a predetermined amount of time.

Referring now to the drawings, and first to FIG. 1, a system according to a preferred embodiment of the present invention is designated generally by reference numeral 10. System 10 is a telephony network system and includes a first public switched telephone network (PSTN) 114a which interfaces to an IP telephony network or Internet 112. The Internet 112 is further interfaced to a second PSTN 114b. The first PSTN 114a includes a source user agent (SUA) 102, i.e., originating agent, which originates a session participation request. The second PSTN 114b includes a called party destination user agent (DUA 103). The Internet 112 includes a redirect server (RS) 104, an SIP proxy server (SPS) 106, a Network Management System (NMS) 108, and destination gateways (DGWs) 110a, 110b. While only two DGWs are shown, one of ordinary skill in the art will recognize that additional DGWs may be provided. The RS 104 supports a gateway management function which tracks the status of the DGWs. The NMS 108 receives and stores all status changes regarding DGWs 110a, 110b. Status changes are reported to the NMS 108 by the RS 104 via the SPS 106. SPS 106 acts as both a server and client for the purpose of making requests on behalf of other clients. SPS 106 provides proxy server and gateway resource management functions. SPS 106 may be a SIP proxy server or an H.323 gatekeeper.

The method of the present invention (i.e., dynamic gateway selection (DGS) and removal) is performed by the SPS 106 and RS 104 in context with the NMS 108 in order to allow calls to be routed to one of the DGWs 110a , 110b. The dynamic gateway selection and removal feature is particularly invoked upon receipt by the RS 104 of a session participation request. The SUA 102 initiates a call attempt to transmit the session participation request to the SPS 106. Accordingly, an attempt is made to establish a communication session between the SUA 102 located in the first PSTN 114a and the called party destination device (DUA) 103 located in the second PSTN 114b. PSTN 114a and 114b are bridged via the Internet 112.

FIG. 2 is a flowchart illustrating the call routing logic for establishing a communication session in accordance with the methodology of the present invention. At step 702, a user initiates a call attempt by sending a session participation request (i.e., an INVITE request) to the SPS 106. When the INVITE request is an initial request, the SPS 106 forwards the initial INVITE request to the RS 104 for routing instructions at step 704. At step 705, it is determined whether there is at least one DGW that can satisfy the request. If so, at step 706, the RS 104 responds to the SPS 106 query with a routing list, i.e., a list of candidate DGWs that can handle the call. The RS 104 supplies the routing list from a gateway information table stored therein.

In the event the RS 104 determines that there are no DGWs that can satisfy the INVITE request, at step 705, a request failure response is returned to the SUA 102, at

step 707. Upon receiving the routing list, the SPS 106 proxies the INVITE request to one of the DGWs 110a, 110b. At step 708, the SPS 106 selects the first DGW 110a in the routing list to determine its serviceability and/or availability status. Steps 710 and 712 determine whether the currently selected DGW 110a from the routing list is in a failure state or has timed out. Specifically, at step 710, a determination is made concerning whether the DGW 110a returns a failure response (i.e., out-of-service response). If there is a failure response at step 710, the SPS 106 reports the gateway failure to the RS 104 at step 714. The RS 104 marks the selected DGW 110a as an out-of-service destination gateway in a gateway information table stored in the RS 104, at step 716.

In addition, the SPS 106 sends a message, (i.e., Simple Network Management Protocol (SNMP) trap), to the NMS 108 indicating a destination gateway failure. The SPS 106 then selects the next DGW 110b in the routing list, at step 718. Control then returns to step 710 to determine the availability of the next selected DGW 110b; that is, whether the next selected gateway 110b is in a failure state or has timed out. If the next selected DGW 110b returns either a failure response at step 710, or times-out at step 712, then steps 714-718 are repeated. In short, the process of selecting a gateway from the routing list and determining whether it is in a failure state or has timed out is repeated until a DGW is found from the routing list which accepts a call with a success response at step 720. When a success response is received, the SPS 106

forwards the response to the calling user SUA 102 at step 722. The media stream for the call is then set up at step 724 to establish a communication link between the SUA 102 and DUA 103.

FIG. 3 is a call routing flow diagram illustrating in greater detail the call routing logic procedure described above with reference to FIG. 2. Table 1 below lists the INVITE required parameter fields in the preferred embodiment when a SUA 102 initiates a call attempt by sending a session participation request (i.e., an INVITE request) to the SPS 106 (See FIG.1, item 1 and FIG. 3, step "a").

Table 1

Parameter	**Usage**
Request-Line	Contains the method (e.g., INVITE), Request Uniform Resource Identifier (i.e., Request-URI) of the SPS and protocol version
To	Contains the address of the recipient of the request
From	Contains the address of the initiator of the request
Call-ID	Uniquely identifies the invitation
Cseq	Contains the request method and a decimal sequence number chosen by the requesting client, unique within a single value of the Call-ID
Via	Indicates the path taken by the request so far

The SIP INVITE is addressed to the called party DUA 103 at a proxy address at the SPS 106. The SIP INVITE specifies the real IP address of the DUA 103. Upon

receipt of the SIP INVITE, the SPS 106 sends a 100 trying message to the ingress or origination gateway (FIG. 3, step "b"). Table 2 lists the mandatory fields associated with the 100 trying response message in the preferred embodiment.

Table 2

Parameter	Usage
Status-Line	Status Code = 100, Reason phrase and protocol version
To	Content copied from the original request message
From	Content copied from the original request message
Call-ID	Content copies from the original request message
Cseq	Content copies from the original request message
Via	Indicates the path taken by the request so far. Add the received-tag parameter if the previous ho address is incorrect in the via header field

The SPS 106 counts the number of INVITE requests received. When a received request message does not contain a route header field, it is determined to be an initial INVITE request message. In such a case, the SPS 106 performs the following steps: (1) if a Topmost Via Header (TVH) source address is incorrect, adds a "Received" parameter (or replaces the existing one if there is one) with the source package to the Via header field inserted by the previous hop; (2) generates an internal Call-ID; or (3) forwards the requested message to the RS 104 (See FIG. 1, item 2 and FIG. 3, step "c"). Table 3 lists the required fields in the RS INVITE request message.

Table 3

Parameter	Usage
Status-Line	Content copied from the original request message
To	Content copied from the original request message
From	Content copied from the original request message
Call-ID	Internally generated Call-I
Cseq	Content copied from the original request message
Via	Add the received tag

In response to receiving the RS INVITE request message from the SPS 106 , the RS 104 performs the following: (1) counts the number of INVITE messages received; (2) determines that the user portion of Request Uniform Resource Identifier (i.e., (Request-URI) is less than or equal to 15 digits and does not contain a leading 0 or 1; and (3) queries a Network Control System/Data Access Point (i.e., NCS/DAP) to obtain routing information (FIG. 3, step "d"). Routing information, including a routing list of prospective DGWs, is then sent from the NCS/DAP to the RS 104 (FIG. 3, step "e"). The routing information is used to update the gateway information table stored in RS 104. The gateway information table is subsequently used to create an updated routing list.

For example, when a gateway address is marked as out-of-service in the gateway information table stored in RS 104 and its associated time value is zero the gateway address is not added to the routing list. When the gateway address is marked

as out-of-service in the gateway information table and its associated time value is greater than the current absolute RS time, the gateway address is not added to the routing list. When the gateway address is marked as out-of-service in the gateway information table and its associated time value is less than or equal to the current absolute RS time, the gateway address is added to the routing list and the gateway address is marked as in-service in the gateway information table. The RS 104 also sends a message, i.e., the Simple Network Management Protocol (SNMP) trap, to the NMS 108 indicating that the DGW is in-service. If there is only one gateway in the routing list, the RS 104 will send a 302 response message back to the SPS 106 (See FIG. 1, item 3 and FIG. 3, step "f"). The RS 104 increments a counter which counts the number of 3xx messages sent. Table 4 lists the required fields in the 3xx (302 in the present case) response message and an example of the contact address list.

Table 4

Parameter	**Usage**
Status-Line	Status Code = 302, Reason phrase and protocol version
To	Content copied from the Original INVITE request
From	Content copied from the Original INVITE request
Call-ID	Content copied from the Original INVITE request
Cseq	Content copied from the Original INVITE request
Via	Content copied from the Original INVITE request
Contact	Multiple gateway addresses

Table 5 lists the required fields in the SNMP trap message to the NMS.

Table 5

Parameter	Usage
Protocol Data Unit (PDU) Type	Indicates that this is a Trap PD
Enterprise	Identifies the network-management subsystem that generated the trap. Its value is taken from sysObjectID in the system Group
Agent-addr	The IP address of the object generating the trap
Generic-trap	6 = enterpriseSpecific. This value signifies that the sending protocol entity recognizes that some enterprise-specific event has occurred. The specific-trap field indicates the type of trap
Specific-trap	A code that indicate more specifically the nature of the trap
Time-stamp	The time between the last (re)initialization of the network entity that issued the trap and the generation of the trap
Variable bindings	The address of the gateway that returned the 5xx response and status (in-service)

In the case where there is more than one gateway in the routing list, the RS 104 sends a 300 response message, instead of a 302 response message for a single gateway, back to the SPS 106. For a 300 response, the RS 104 also counts the number of 3xx responses sent. Table 6 lists the required fields in the 300 response message and an example of the contact address list.

Table 6

Parameter	Usage
Status-Line	Status Code = 300, Reason phrase and protocol version
To	Same as the original INVITE
From	Content copied from the Original INVITE request
Call-ID	Content copied from the Original INVITE request
Cseq	Content copied from the Original INVITE request
Via	Content copied from the Original INVITE request
Contact	Multiple reachable addresses

The SPS 106 counts the number of routing responses received from the RS 104. The SPS 106 sends an INVITE to the first DGW 110a, and inserts the “user=phone” header in each contact list address (See FIG. 1, item 4 and FIG. 3, step “g”). Table 7 lists the required fields of the INVITE request sent from the SPS 106 to the DGW 110a.

Table 7

Parameter	Usage
Request-Line	Contains the method (e.g., INVITE), Request-URI using the gateway from the top of the unused contact list and protocol version

To	Content copied from the original request message
From	Content copied from the original request message
Call-ID	Content copied from the original request message
Cseq	Content copied from the original request message
Via	Add the NS URL to the top
Record Route	Request-URI of the NS

The SPS 106 may receive a 100 trying response (see FIG. 3, step "h") or a 180 ringing response from the DGW 110a (See FIG. 3, step "i"). Table 8 lists the required fields of the 180 ringing response.

Table 8

Parameter	**Usage**
Status-Line	Status Code = 180, Reason phrase and protocol version
To	Same as the original INVITE request plus tag
From	Same as the INVITE request sent to the Egress Gateway
Call-ID	Same as the INVITE request sent to the Egress Gateway
Cseq	Same as the INVITE request sent to the Egress Gateway
Via	Same as the INVITE request sent to the Egress Gateway

After receiving the 180 ringing response from the DGW 110a, the SPS 106 removes itself from the top of the Via field, re-starts the invite User Agent (UA) timer if it exists, and forwards the 180 ringing response to the ingress gateway (See FIG. 3, step "j"). The response message is sent to the address indicated in the Via header field. Table 9 lists the required message elements.

Table 9

Parameter	**Usage**
Status Line	Content copied from the 180 response received from the gateway
To	Content copied from the 180 response received from the gateway
From	Content copied from the 180 response received from the gateway
Call-ID	Content copied from the 180 response received from the gateway
Cseq	Content copied from the 180 response received from the gateway
Via	Content copied from the 180 response received with the removal of the NS URL

The SPS receives an INVITE response (i.e., 200 OK response) from the DGW 110a (See FIG. 3, step "k"). Table 10 lists the required fields in the INVITE message.

Table 10

Parameter	Usage
Status Line	Status Code = 200, Reason phrase and protocol version
To	Same as the original INVITE request plus tag
From	Same as the INVITE request sent to the Egress Gateway
Call-ID	Same as the INVITE request sent to the Egress Gateway
Cseq	Same as the INVITE request sent to the Egress Gateway
Record Route	Request-URI of the NS
Contact	The reachable address of the Egress Gateway

After receiving the 200 OK response from the DGW 110a, the SPS 106 performs the following steps: (1) cancels the invite UA timer, if it exists; (2) removes the SPS URL from the Topmost Via Header (TMVH) field; (3) adds the next hop's Request-URI at the top of the record-route header field when either of the following conditions are met: a) there is no contact header field in the 200 OK response, or b) the SPS URL is on the top entry of the record-route header field; (4) counts the number of 200 INVITE responses sent by the SPS 106; and (5) the SPS 106 starts the ACK timer. The SPS 106 forwards the INVITE response (i.e., 200 OK response) to the ingress gateway (See FIG. 3, step "I"). Table 11 lists the required headers in the INVITE response.

Table 11

Parameter	Usage
Status-Line	Status Code = 200, Reason phrase and protocol version
To	Same as the original INVITE request plus tag
From	Same as the INVITE request sent to the Egress Gateway
Call-ID	Same as the INVITE request sent to the Egress Gateway
Cseq	Same as the INVITE request sent to the Egress Gateway
Via	Content from the INVITE request sent to the Egress Gateway
Record Route	Request-URI of the NS
Contact	The reachable address of the Egress Gateway

The SPS 106 receives an ACK response from the ingress gateway and stops the ACK timer. The SPS 106 counts the number of ACK response messages received by the SPS 106. Table 12 lists the required headers of the ACK response (See Fig. 3, step "m").

Table 12

Parameter	Usage
Request-Line	Contains method (e.g., ACK), Request-URI of the NS and protocol version
To	Same as the original INVITE plus the tag
From	Same as the original INVITE
Call-ID	Same as the original INVITE
Cseq	Same sequence number as the original INVITE

Via	UA originated

The received ACK response may contain a route header field. The SPS 106 either proxies the ACK response using the address in the route header or uses the address in the "To" header to determine whether to proxy the ACK response or consume the ACK response. The ACK response will be consumed when the "To" header address is equal to the SPS address, and no route header exists in the ACK response. When the SPS 106 determines that the ACK response should be proxied, the SPS 106 performs the following: (1) the SPS 106 adds the ACK's address to the top of the Via field; (2) the SPS 106 removes the top address from the route header field; (3) the Request-URI is set to the address located at the top of the route header field; and (4) the ACK message is forwarded to the DGW 110a based on the top address in the route header field if it exists or based on the call context's DGW information (See FIG. 3, step "n"). Accordingly, the DGW 110a is selected as the available gateway for completing the call. If the DGW 110a is determined to be unavailable, the same method outlined above is used to determine if DGW 110b is available. If neither DGW is available, a message is sent to the SUA 102 that the call cannot be completed. Table 13 lists the parameters of the ACK request message sent to the DGW 110a.

Table 13

Parameter	Usage

Request-Line	Contains method (e.g., ACK), Request-URI is copied from the top list of Route header field and protocol version
To	Content copied from the ACK received from the Ingress Gateway
From	Content copied from the ACK received from the Ingress Gateway
Call-ID	Content copied from the ACK received from the Ingress Gateway
Cseq	Content copied from the ACK received from the Ingress Gateway
Via	UA originated with the NS URL added to the top of Via field

In another preferred method, a DGW is selected using a ping method. In this embodiment, gateway availability is determined by sending at least one packet to each destination gateway to ascertain whether the destination gateway is available, or in-service. Accordingly, if the destination gateway is in-service, it transmits an ACK message. If an ACK message is not received after a predetermined period of time, the DGW is determined to be unavailable. The ping method preferably queries each destination gateway one-by-one and updates a gateway information table by recording the status of each gateway. For example, if the ACK message is received, the DGW is then checked to determine if it is available. If it is available, its address is stored in a gateway information table, and the process repeats for the next DGW.

FIG. 4 is a flowchart illustrating the methodology of an alternate embodiment of the present invention, i.e., the ping method. A counter is initialized at step 800 to indicate the currently selected destination gateway from the routing list (i.e., i=1 to the number of gateways in the routing list). In step 802, a message is transmitted from a server (e.g. proxy server, redirect server) to the ith destination gateway for the purpose of obtaining a response. In step 804 the server awaits a response from the ith destination gateway. Step 806 is a determination step to determine whether a response was received from the ith destination gateway. If a response is not received within a predetermined amount of time, the process continues at step 808 where it is determined whether there are additional destination gateways to check from the routing list. If not, the process terminates at step 810. Otherwise, the counter is incremented to select a succeeding destination gateway from the routing list at step 812. Steps 802-806 are then repeated to check the response and/or availability of the succeeding (i.e. ith + 1) destination gateway from the routing list. If it is determined at step 806 that a response is received within the predetermined time, the process continues to step 814 where it is then further determined whether the responding destination gateway is available or not. If the destination gateway is not available, the process returns to step 808 to determine whether there are additional destination gateways to check from the routing list. If so, steps 802-806 are repeated for the succeeding destination gateway. Otherwise, if it is determined at step 814 that the responding destination gateway is

available, the process continues at step 816 where the address of the available destination gateway is stored in a gateway information table. The process returns to step 808 to determine if there are additional destination gateways to be checked in the routing list.

What has been described herein is merely illustrative of the application of the principles of the present invention. For example, the functions described above for operating the present invention are for illustration purposes only. Other arrangements and methods may be implemented by those skilled in the art without departing from the scope and spirit of the invention.